18 Signal Processing in Process Control and Automation

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INTRODUCTION

Signals are processed for a variety of reasons, such as to remove unwanted noise, to correct distortion, to make it suitable for transmission, or to extract certain meaningful information. In this chapter, the mathematical foundations of analog and digital signal processing will be given. Some examples of practical hardware implementation will also be introduced.

A majority of process control and automation equipment on the field such as the sensors and transducers and the control elements generate or operate on analog signals. At the same time, it is very likely that the entire process is built on digital technology. Hence, there is a wide range of signal-processing techniques need to be considered in instrumentation and process control applications. Although signal processing is a very broad topic, however, we will view signal processing in three categories:

1. Signal processing for sensor, transducer, and controllers
2. Signal processing for communication purposes
3. Signal processing for networks

In this chapter, we will concentrate mainly on the sensor and transducer signals. There are virtually hundreds and thousands of different sensors and transducers used in process control and automation. All these sensors and transducers generate different types of signals that require different of signal-processing techniques, some of which will be explained here.

The sensors and transducers can broadly be classified as follows:

- Voltage sensor/transducer: examples are electromagnetic (EM) transducers, tachogenerators, and rotary transformers. They require high-input impedance amplifiers.
- Current and charge sensor/transducer: examples are piezoelectric, pyroelectric sensors, and semiconductor detectors. They are sensitive to capacitances in the amplifier circuits.
- Resistive sensor/transducer: examples are strain gages, pressure, force, acceleration sensors, and temperature sensors. They often use bridge methods for signal processing and high-precision components must be selected.

A majority of sensor and transducer signals are analogue in nature and will be explained next.

ANALOG SIGNALS: MATHEMATICAL APPROACH

An analog signal is a physical quantity that conveys information about some physical phenomenon in a continuous nature. Common physical quantities include electrical charges, voltages, currents, and frequencies whereas physical phenomena can be temperature, position, sound, vibration, light intensity, and so on. For example, an analog signal can be a voltage that may vary in amplitude, frequency, or phase in response to changes in temperature.

An analog signal is represented by a continuous function of two or more independent variables. The most commonly used variable is the time. Hence, the term *continuous-time* is often used to describe analog signals. This article will deal with one-dimensional (ID) analog signals that are a function of time, which can be described by mathematical expressions or graphical plots (Figure 18.1).

Analog Systems

Analog instruments operate on purely analog principle, that is, they measure, transmit, display, and store data in analog form. Analog instruments require the following considerations:

- Signal transmission and conditioning
- Loading effects, buffering, and impedance matching
- Shielding
- Inherent and imposed noises
- Signal level compatibility
- Data storage

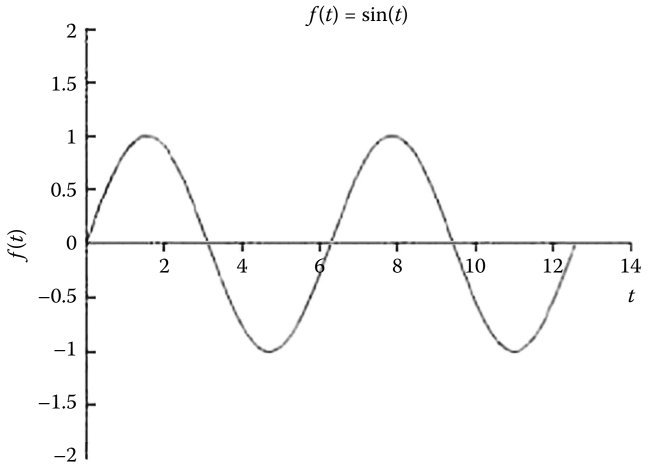


FIG. 18.1
Example of an analog signal.

The signal conditioning is usually realized by integrating many functional blocks such as bridges, amplifiers, filters, oscillators, modulators, offset circuits, level converters, buffers, and the like, as in Figure 18.2.

An analog system is a collection and interconnection of several building blocks, which performs some operation on an input signal to produce an output signal. The input signal is known as the excitation signal, while the output or the processed signal is referred to as the response signal. In general, a system may have multiple inputs and outputs.

Figure 18.3 depicts a block diagram representation of an analog system H that accepts a single excitation signal $x(t)$ and then produces a single response signal $y(t)$.

The relationship between $x(t)$ and $y(t)$ is governed by the system H transfer function. Mathematically, it can be expressed as

$$y(t) = H[x(t)] \tag{18.1}$$

Analog systems can be classified as linear or nonlinear, causal or noncausal, time invariant or time varying, stable or non-stable. Among them, systems that are linear and time invariant (LTI) have special significance for the purpose of mathematical analysis.

Random Signals

Random signals appear in time in an unpredictable manner. They cannot deterministically be described to any reasonable degree of accuracy or description is too complicated for practical use. Theoretically, an infinitely long time record is necessary to obtain a complete description of random signals. However, statistical methods and probability theory can be used for the analysis by taking representative samples. Mathematical tools such as probability distributions, probability densities, frequency spectra, cross-correlations, auto-correlations, digital fourier transforms (DFTs), fast fourier transforms (FFTs), and auto-spectral analysis, rms values, and digital filter analysis are some of the techniques that can be employed.

Periodic/Aperiodic Signals

A periodic signal repeats itself at regular intervals. In general, any signal $x(t)$ for which is

$$x(t) = x(t - T) \tag{18.2}$$

for all t is said to be *periodic*. The fundamental period of the signal is the minimum positive, nonzero value of T for which Equation 18.2 is satisfied. If a signal is not periodic, then it is *aperiodic*.

Symmetric Signals

There are two types of signal symmetry: odd and even. A signal $x(t)$ has *odd symmetry* if and only if

$$x(-t) = -x(t) \tag{18.3}$$

for all t. It has *even symmetry* if and only if

$$x(-t) = x(t) \tag{18.4}$$

Continuous/Discrete Time Signals

A continuous-time signal is defined for all values of t. A discrete time signal is only defined for discrete values of $t = \ldots, t_{-1}, t_0, t_1, \ldots, t_n, t_{n+1}, t_{n+2}, \ldots$ It is uncommon for the spacing between t_n and t_{n+1} to be change with n. The spacing is most often some constant value referred to as the sampling rate:

$$T_s = t_{n+1} - t_n \tag{18.5}$$

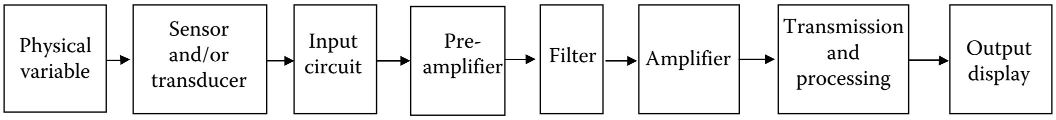


FIG. 18.2
Block diagram of an analog instrument.

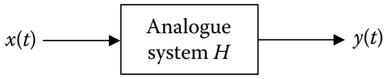


FIG. 18.3
Block diagram of an analog system.

It is often convenient to express discrete time signals as

$$x(nT_s) = x[n] \tag{18.6}$$

That is, if $x(t)$ is a continuous-time signal, then $x[n]$ can be considered as the nth sample of $x(t)$.

Sampling of a continuous-time signal $x(t)$ to yield the discrete-time signal $x[n]$ is an important step in the process of digitizing the signal.

Signal Energy and Power

When the strength of a signal is measured, it is usually the signal power or signal energy that is of interest. The signal power of $x(t)$ is defined as

$$P_x = \lim_{T\to\infty} \frac{1}{T} \int_{-T/2}^{T/2} x^2(t)dt \tag{18.7}$$

and the signal energy as

$$E_x = \lim_{T\to\infty} \int_{-T/2}^{T/2} x^2(t)dt \tag{18.8}$$

A signal for which P_x is finite and nonzero is known as a *power signal.* A signal for which E_x is finite and nonzero is known as an *energy signal.* P_x is also known as the *mean-square* value of the signal.

Signal power is often expressed in the units of decibels (dB). The decibel is defined as

$$P_{x\mathrm{dB}} = 10\log\left(\frac{P_x}{P_0}\right) \tag{18.9}$$

where P_0 is a reference power level, usually equal to one squared SI unit of the signal. For example, if the signal is a voltage, then the P_0 is equal to one square Volt.

Signal-to-Noise Ratio

Any measurement of a signal necessarily contains some random noise in addition to the signal. In the case of additive noise, the measurement is

$$x(t) = s(t) + n(t) \tag{18.10}$$

where

$s(t)$ is the signal component
$n(t)$ is the noise component

The signal-to-noise ratio (SNR) is defined as

$$SNR_x = \left(\frac{P_s}{P_n}\right) \tag{18.11}$$

or in decibels

$$SNR_x = 10\log\left(\frac{P_s}{P_n}\right)\mathrm{dB} \tag{18.12}$$

The SNR is an indication of how much noise is contained in a measurement.

Impulse Response

Any LTI system may be represented in the time domain by its response to a specific signal called the *impulse.* A unit impulse, also known as the Dirac delta function $\delta(t)$, is defined as

$$\delta(t) = \begin{cases} \infty, & t = 0, \\ 0, & t \neq 0. \end{cases} \tag{18.13}$$

The unit impulse is zero everywhere except at the origin where it is infinitesimally narrow while its amplitude is infinite. Supposing $h(t)$, as shown in Figure 18.4, is the response of an analog LTI system H to a unit impulse $\delta(t)$.

Then, by knowing $h(t)$, the behavior of the system H with input signal $x(t)$ and output signal $y(t)$ can be described as

$$y(t) = \int_{-\infty}^{\infty} x(\tau)h(t-\tau)d\tau \tag{18.14}$$

This equation is known as the *convolution integral.* It shows that the knowledge of the impulse response $h(t)$ is a complete characterization of the LTI system. Equation 18.14 can also be written by using the convolution operator * as

$$y(t) = x(t) * h(t) \tag{18.15}$$

Note that in practical system design, the causality and stability of the system are required. A system is considered causal if and only if the current output does not depend on any future values of the input. For physically realizable LTI systems, a necessary and sufficient condition for causality is that its impulse response be zero prior to application of the excitation.

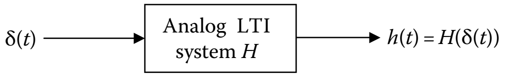


FIG. 18.4
Impulse response.

Differential Equation Representation

Although LTI systems are completely characterized by their impulse response, in practice they are often described by linear constant-coefficient differential equations.

An analog LTI system with excitation signal $x(t)$ and response signal $y(t)$ can be modeled using a linear constant-coefficient ordinary differential equation (ODE):

$$\begin{aligned}\frac{d^n y(t)}{dt^n} + a_{n-1}\frac{d^{n-1}y(t)}{dt^{n-1}} + \cdots + a_1\frac{dy(t)}{dt} + a_0 y(t)\\ = b_m\frac{d^m x(t)}{dt^m} + b_{m-1}\frac{d^{m-1}x(t)}{dt^{m-1}} + \cdots + b_0 x(t)\end{aligned} \qquad \textbf{(18.16)}$$

where a_k and b_k are coefficients. The response of a system can be obtained by solving the above equation.

Laplace Transform

The Laplace transform is a technique for solving ODEs with constant coefficients. It converts a set of differential equations into a corresponding set of algebraic equations, which is much easier to solve. These algebraic equations are no longer functions of time but of complex frequency variable. Once they are solved to give the Laplace transform of the desired form, the inverse Laplace transform is carried out to find the corresponding solution as a function of time.

The unilateral (or one-sided) Laplace transform of a signal $x(t)$ is defined as

$$X(s) = \int_0^\infty x(t)\mathrm{e}^{-st}dt \qquad \textbf{(18.17)}$$

where s is a complex frequency variable. It is unilateral as it has no information about the signal for $t<0$. If it is desired to include the behavior of the signal for negative time then the bilateral Laplace transform can be used instead. The bilateral (or double-sided) Laplace transform is defined as

$$X_B(s) = \int_{-\infty}^\infty x(t)\mathrm{e}^{-st}dt \qquad \textbf{(18.18)}$$

The time function $x(t)$ and its Laplace transform $X(s)$ constitute a unique Laplace transform pair. The inverse Laplace transform can be evaluated by decomposing $X(s)$ into the sum of simpler terms (employing partial fraction expansion if necessary) such that each has an inverse transform, which can be easily recognized from a stock list of Laplace transform pairs.

Laplace transform has a number of important operational properties, one of which states that the convolution in the time domain corresponds to multiplication in the frequency (or s) domain. Therefore, the s-domain equivalent of the convolution is

$$Y(s) = X(s)H(s) \qquad \textbf{(18.19)}$$

Assuming that the system is relaxed (i.e., all initial conditions are zero), $H(s)$ is the *transfer function* of the relaxed LTI system, which is defined as

$$H(s) = \left(\frac{Y(s)}{X(s)}\right) \qquad \textbf{(18.20)}$$

It is the ratio of the Laplace transform of the output $y(t)$ to the Laplace transform of the input $x(t)$. The transfer function $H(s)$ can also be defined as the Laplace transform of the system impulse response $h(t)$.

By assuming zero initial conditions and applying the derivative property of the Laplace transform to the differential equation representation as stated in Equation 18.20, the system transfer function $H(s)$ can be expressed as

$$H(s) = \frac{b_m s^m + b_{m-1}s^{m-1} + \cdots + b_1 s + b_0}{s^n + a_{n-1}s^{n-1} + \cdots + a_1 s + a_0} \qquad \textbf{(18.21)}$$

This shows that the system transfer function is a ratio of polynomials in s. The above equation may be expressed in factored form as

$$H(s) = K\frac{(s-z_1)(s-z_2)\cdots(s-z_m)}{(s-p_1)(s-p_2)\cdots(s-p_n)} \qquad \textbf{(18.22)}$$

where K is a constant multiplier. The roots of the numerator, $z_1, z_2, \ldots, z_m$, are called the *zeros* of the system, and they are the values of s for which the transfer function $H(s)$ is zero. The roots of the denominator, $p_1, p_2, \ldots, p_n$, are called the *poles* of the system, and they are the values of s for which $H(s)$ equals infinity.

These root locations can be visualized by plotting the poles as crosses “x” and zeros as circles “o” on the s-plane. This plot is referred to as the *pole-zero plot*.

Fourier Transform

The Fourier transform of an analog signal $x(t)$ is defined as

$$X(f) = \int_{-\infty}^\infty x(t)\mathrm{e}^{-j\omega t}dt \qquad \textbf{(18.23)}$$

in terms of the frequency variable ω.

The inverse Fourier transform is defined as

$$x(t) = \frac{1}{2\pi}\int_{-\infty}^\infty X(f)\mathrm{e}^{j\omega t}d\omega \qquad \textbf{(18.24)}$$

Laplace and Fourier transforms are closely related. In Laplace transform, the complex frequency variable *s* generalizes the concept of frequency to the complex domain, and is defined as

$$s = \sigma + j\omega \tag{18.25}$$

where
σ represents the real part of *s*, and
ω (which equals to 2π*f*) represents the imaginary part of *s*

If the signal *x*(*t*) is causal and absolutely integrable, its Fourier transform *X*(*f*) corresponds to the Laplace transform *X*(*s*) that is evaluated on the *j*ω axis (i.e., $s = 0 + j2\pi f$).

Frequency Response

A fundamental property of LTI systems is that the complex exponential signal $x(t) = e^{j\omega t}$ is an *Eigenfunction*, that is, the response of any LTI system to $e^{j\omega t}$ is

$$y(t) = H(j\omega)e^{j\omega t} \tag{18.26}$$

where *H*(*j*ω) is the Fourier transform of the system impulse response:

$$H(j\omega) = \int_{-\infty}^{\infty} h(t)e^{-j\omega t}dt \tag{18.27}$$

The *Eigenvalue H*(*j*ω) is called the *frequency response* of the system. It carries the same information as the impulse response, and, hence, either function can be used to characterize the system. Another interpretation of the frequency response is that it is the ratio of the Fourier transform of the output signal to the Fourier transform of the input signal:

$$H(j\omega) = \left(\frac{Y(j\omega)}{X(j\omega)}\right) \tag{18.28}$$

This result is a direct consequence of the convolution theorem which states that convolution in the time domain corresponds to multiplication in the frequency domain. For causal systems, the frequency response can be obtained by evaluating the system transfer function, *H*(*s*), on the *j*ω axis. In general, *H*(*j*ω) can be determined by the input–output measurement. However, if the system is described by an ODE, then *H*(*j*ω) is given by

$$H(j\omega) = \frac{b_m(j\omega)^m + b_{m-1}(j\omega)^{m-1} + \cdots + b_1 j\omega + b_0}{(j\omega)^n + a_{n-1}(j\omega)^{n-1} + \cdots + a_1 j\omega + a_0} \tag{18.29}$$

Magnitude and Phase Representation of the Frequency Response

The frequency response is a complex quantity, which describes the effect of the filter on the frequency components of the input signal. It can be represented by two real quantities, namely, the magnitude and phase functions:

$$H(j\omega) = |H(j\omega)|\angle H(j\omega) \tag{18.30}$$

$$\text{Magnitude: } |H(j\omega)| = \sqrt{\mathrm{Re}\{H(j\omega)\}^2 + \mathrm{Im}\{H(j\omega)\}^2} \tag{18.31}$$

$$\text{Phase: } \angle H(j\omega) = \arg\{H(j\omega)\} = \tan^{-1}\left\{\frac{\mathrm{Im}\{H(j\omega)\}}{\mathrm{Re}\{H(j\omega)\}}\right\} \tag{18.32}$$

where Re{·} and Im{·} denote, respectively, the real and imaginary parts of the argument. The magnitude and phase functions are also called the *magnitude response* and *phase response*. One important property of the frequency response of real systems is conjugate symmetry:

$$H(j\omega) = H^*(-j\omega) \tag{18.33}$$

It follows from this property that the magnitude response is an even function and the phase response is an odd function, that is,

$$|H(j\omega)| = |H(-j\omega)| \quad \text{and} \quad \angle H(j\omega) = -\angle H(-j\omega) \tag{18.34}$$

This is why it is common to specify the magnitude and phase responses for positive ω only.

ANALOG SIGNAL PROCESSING: DESIGN APPROACH

The term analog signal processing describes a body of techniques that can be implemented to process analog (or real-world) signals. This includes the theory and application of filtering, coding, transmitting, estimating, detecting, analyzing, and reproducing analog signals. In today's digital world, analog signal processing plays a fundamental role. It converts real-world information such as voice, sound, pressure, or voltage into the 1s and 0s of the digital domain so that it can be processed by microprocessors, computers, microcontrollers, and digital signal processors (DSPs). The conversion from 1s and 0s back into the real-world signals requires again analog signal processing. Without the aid of analog signal processing, digital circuits would not be able to process real-world signals. As a result, most electronic equipment comprises analog processing circuitry that acts as interface between the digital world and the real world.

Basic Circuit Components

The three basic electronic components are the resistor, capacitor, and inductor. The resistor's function is to limit the current, the greater the resistance value *R* the smaller the current flowing through the resistor for the same voltage. In contrast, a current will flow through a capacitor only as a result of a change in the voltage across it. Consequently, a capacitor can be used to block direct current (DC) and pass alternating current (AC). In the case of an inductor, voltage is established only as a result of a change in the current across it. Another electronic component is the diode, a semiconductor device that allows current to flow through it in only one direction. Diodes can thus be used as a rectifier in a power supply device to convert AC into DC. They can also be used as an on/off switch that controls current.

Other basic electronic components are the transistors. Their function is to provide gain, that is, amplify an electric current. There are many different kinds of transistors, the most encountered being the bipolar and MOSFET transistors. All types of transistors provide gain but differ by their properties, making them, for instance, either well or ill suited for high-speed applications. An integrated circuit (IC) contains all basic circuit components from resistors to transistors, densely packed on a silicon chip. The operational amplifier (commonly known as op amp) is an example of IC. It is a fundamental building block for analog signal processing. It provides more gain than a single transistor and allows an easy implementation of a wide variety of computational building blocks for analog signal processing.

The properties of some basic circuit components are summarized in the Table 18.1.

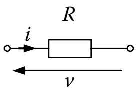
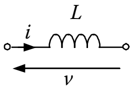
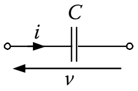
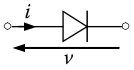
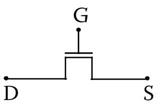
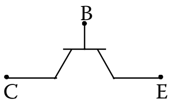
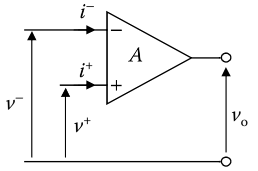
Operational Amplifiers

Operational amplifiers are the building blocks of instruments. The standard operational amplifier is shown in Figure 18.5, it has two input terminals called inverting inputs (−) and non-inverting input (+), and one output terminal. Performance of op-amps depends on offset voltages, offset currents, dynamic loop gains, Common mode rejection ratios (CMRRs), power supply rejection ratios, slew rates, and output ratios.

In order to get a concept, we will consider its ideal characteristics. A practical op-amp, however, falls short of these ideal standards. The ideal op-amp has infinite voltage gain and infinite bandwidth. Also, it has an infinite input

TABLE 18.1

Some Basic Electronic Components

Components	*Symbol*	*Property*	*Characteristics*
Resistor		$v = Ri$	Limits the current
Inductor		$v = L\,(di/dt)$	Voltage established as a result of a change in current
Capacitor		$i = C\,(dv/dt)$	Passes AC and blocks DC
Diode		$i = I_s(e^{u/u_T} - 1)$	Allows current to flow through it in only one direction
MOSFET transistor		Provides amplification	The voltage on the Gate (G) controls the flow of electrons from the Source (S) to the Drain (D), low input impedance
Bipolar transistor		Provides amplification	A small Base–Emitter current controls the larger collector–emitter current, high-input impedance
Operational amplifier		Provides amplification $v_o = A(v^+ - v^-)$	High-input impedance > 10^6 Ω Low-output impedance < 200 Ω High amplifier gain A > 10^4

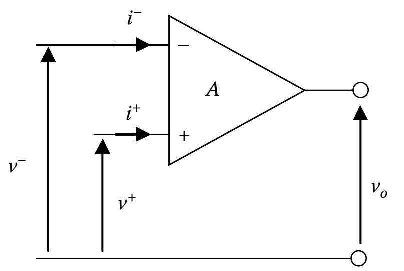


FIG. 18.5
Representation of an operational amplifier.

impedance (open), so that is doesn't draw any power from the driving source. Finally, it has zero output impedance. A good practical operational amplifier has the following properties:

- High input resistance, hundreds of megaohms or few gigaohms
- Low output resistance, less than few ohms or fraction of one ohm
- Low input offset voltage, few millivolts or in microohms
- Low input bias current, few picoamps
- Low intrinsic noise
- High common mode rejection ratio
- Low sensitivity to changes in power voltage
- A broad operating frequency range and bandwidth

Op-amps are used as common amplifiers as in voltage followers, limiters, oscillators, comparators, drivers, voltage-to-frequency converters, frequency-to-voltage converters, differentiators, integrators, summers, multipliers/dividers, filters, buffers, modulators/demodulators, function generators, linearizers, and so on. Only few of these configurations will be explained next.

Amplifiers

Sensors and transducers generally produce signals with low amplitudes. Therefore, noise-free amplifications of transducer signals are an essential part of instrumentation. An amplifier provides amplification of the input signal in terms of voltage, current, or power. There are many types of amplifiers and many ways to classify them. For instance, they can be classified as voltage or power amplifiers depending on whether it is the input voltage or input power that is amplified or they can be classified according to the frequency band across which they can operate: audio amplifiers (15 Hz–20 kHz), RF amplifiers (20 kHz–0.1 GHz), and video amplifiers (10 Hz–6 MHz). Depending on the nature of the input/output (I/O) signal (voltage or current), they can also be classified in four categories: voltage amplifiers (input and output are voltages), current amplifiers (input and output are currents), transconductance amplifiers (input voltage and output current), or transresistance amplifiers (input current and output voltage). In all cases and irrespective of the adopted designation, the essential function of the amplifier is to achieve signal amplification, which enables the processing and transmission of "weak" (i.e., small magnitude) signals.

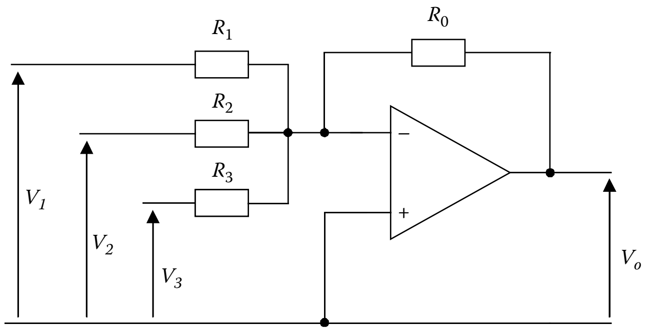


FIG. 18.6
A three-input adder op-amp circuit.

Adders and Multipliers

An adder is a circuit whose output is the weighted algebraic sum of all its inputs. In the case of the voltage adder of Figure 18.6 with inputs V_1, V_2, and V_3, the output will be inverted as

$$V_0 = -\frac{R_0}{R_1}V_1 - \frac{R_0}{R_1}V_2 - \frac{R_0}{R_1}V_3 \qquad \textbf{(18.35)}$$

A multiplier can be defined as a circuit whose output is proportional to the product of its inputs. In the case of a voltage multiplier with inputs V_1 and V_2, the output will be

$$V_o = kV_1V_2 \qquad \textbf{(18.36)}$$

where k is a constant with suitable dimension. If the multiplier can accept inputs of either polarity, it will be referred as a four-quadrant multiplier. For a two-quadrant multiplier, only one of the two inputs can be bipolar. Finally, a one-quadrant multiplier can accept only unipolar inputs. Multipliers can be realized using the basic components.

Integrators and Differentiators

An integrator can be defined as a circuit whose output rate of change is proportional to the applied input. In the case of an integrator with input V_i, the output will be

$$\frac{dV_0}{dt} = kV_i \quad \text{or} \quad V_0 = k\int V_i dt \qquad \textbf{(18.37)}$$

This equation shows that the output V_o is obtained by integration of the input V_i, hence the name integrator. The latter can be used to time-average a signal, determine its dc level or reduce noise in some cases. Figure 18.7 illustrates how an integrator can be implemented using an op-amp. The

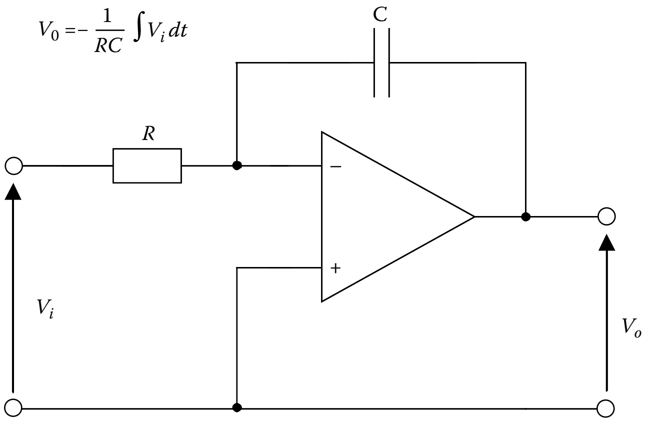


FIG. 18.7
Integrator op-amp circuit.

integrator is an important computational building block in the design of active filters in particular.

A differentiator can be defined as a circuit whose output is proportional to the input rate of change. In the case of a differentiator with input V_i, the output will be

$$V_0 = k\frac{dV_i}{dt} \tag{18.38}$$

This equation shows that the output V_o is obtained by differentiation of the input V_i, hence the name differentiator. Differentiators can be implemented using an op-amp.

Current-to-Voltage and Voltage-to-Current Converters

In the previous sections, it has been assumed that input signals were voltages. However, I/O signals can be either voltage or currents. Current-to-voltage (I–V) and voltage-to-current (V–I) converters can be used to transform signals in the most suitable form for measurement or processing. Figure 18.8 shows how such converters can be implemented using op-amp circuits. In the case of a V–I converter, the output is a current I_0 proportional to the input voltage V_i. Note that the portion of circuit between points A and B acts as a current source controlled by the input voltage V_i. In the case of the I–V converter, the output is a voltage V_0 proportional to the input current I_i. Note that whatever the load resistor placed at the output, the entire input current I_i will flow through resistor R_1. Thus, the portion of the circuit between point S and ground acts as a voltage source controlled by the input current I_i.

Modulators and Demodulators

In electronic communication systems, modulation and demodulation are essential functions that allow convenient and efficient transmission and reception of information and contribute to the sharing of the limited EM spectrum. Modulation is the process of encoding information on a carrier signal such that its amplitude, frequency, or its phase varies with the message signal, referred to as the modulating signal. There are three types of modulation, namely; amplitude modulation (AM), frequency modulation (FM), and phase modulation. All other types of modulation, such as pulse modulation, can be considered as belonging to one of these three modulation types. Examples of AM and FM signals are illustrated in Figure 18.9. Demodulation is the process by which information is extracted back from the carrier signal.

The terms **modulators** and **demodulators** refer to devices that carry out the processes of modulation and demodulation, respectively. Amplitude modulation can be achieved using the technique depicted in Figure 18.10a. It simply requires a multiplier together with an oscillator to establish the frequency of the carrier signal. At the receiver end, the AM signal can be demodulated using a multiplier along with a low-pass filter as depicted in Figure 18.10b. The latter technique is called coherent or synchronous demodulation. Envelope detection is another widely adopted demodulation technique chosen for its simple circuit implementation.

A frequency modulator is an oscillator whose frequency can be changed in accordance with the signal carrying information. A simple example of FM modulator is an LC (inductive-capacitive) oscillator using a voltage-variable capacitor (varactor) whose capacitance exhibits a linear

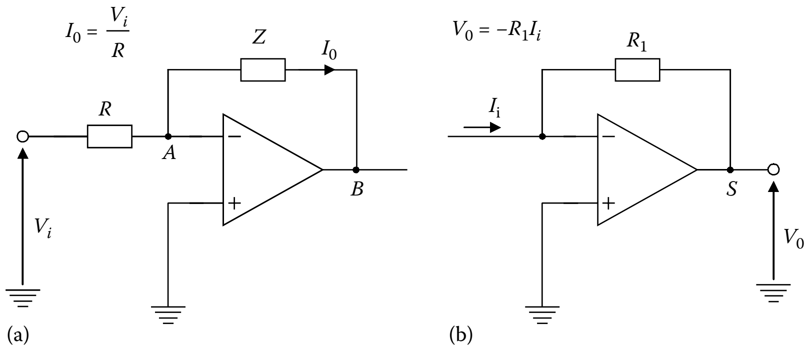


FIG. 18.8
(a) V–I converter and (b) I–V converter.

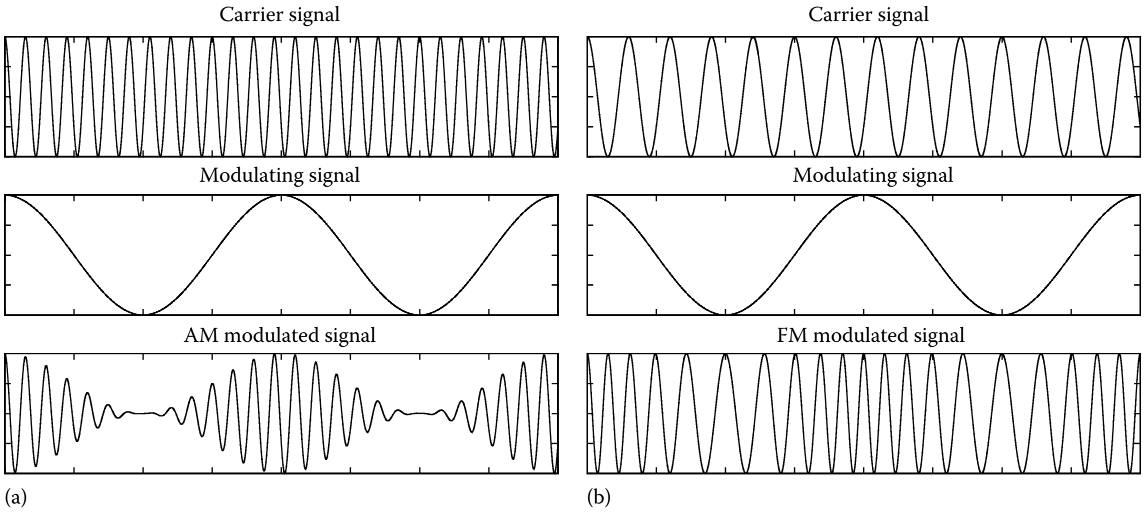


FIG. 18.9
(a) AM modulation and (b) FM modulation.

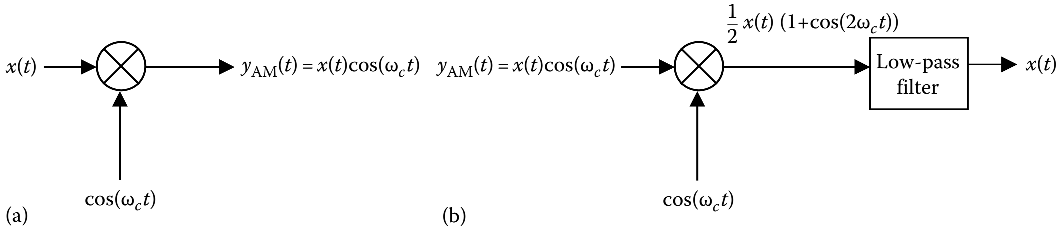

$x(t)$
$y_{AM}(t) = x(t)\cos(\omega_c t)$
$\cos(\omega_c t)$
(a)
$y_{AM}(t) = x(t)\cos(\omega_c t)$
$\frac{1}{2}x(t)\ (1+\cos(2\omega_c t))$
Low-pass filter
$x(t)$
$\cos(\omega_c t)$
(b)

FIG. 18.10
A block diagram of (a) amplitude modulation and (b) demodulation.

dependence with the change in the applied voltage. The frequency of the LC oscillator can thus be controlled by the modulating signal. This type of variable reactance modulator is also called a voltage controlled oscillator (VCO). At the receiver end, the FM signal will be demodulated by filtering.

ANOLOG FILTERS

Filters constitute a very important section of analog signal processing. Therefore, they will be dealt with in a greater detail here as a typical example of analog signal processing.

Filtering is the process by which the essential information, called the signal, is separated from the nonessential information, called *noise* or *interference*. A filter is a system that can modify or reshape the frequency spectrum according to some prescribed specifications. For example, it may amplify a certain frequency component and attenuate others, or it may select a frequency component to pass through it and block others, and so on. Electric filters are widely used in modern electronics, for example:

- To extract useful signals from the sensors
- To obtain DC voltages in power supplies
- To eliminate contamination and noise in communication systems
- To separate radio and television signals
- To improve the quality of audio equipment

Analog filters can be classified based on the physical components they are constructed from:

- Passive RLC filters
- Active RC (resistive capacitive) filters
- Switched-capacitor filters
- Microwave filters
- Crystal filters

Although there are nonlinear filters, filtering is most commonly achieved using LTI systems. By the convolution property of the Fourier transform, the output of an analog filter, in the frequency domain, is given by

$$Y(j\omega) = H(j\omega)X(j\omega) \tag{18.39}$$

Clearly, if at a certain frequency $H(j\omega) = 1$, that frequency component passes unattenuated, whereas if at a certain frequency $H(j\omega)=0$ that frequency component is rejected, that is, filtered out of the input signal.

Frequency-Selective Filters

Frequency-selective filters are systems that select only a range of frequencies to pass through and stop or reject the other frequencies. They can be classified, according to the function they are to perform, as *low-pass*, *high-pass*, *band-pass*, or *bandstop* filters:

- A low-pass filter passes low frequencies from DC ($\omega=0$) to a cutoff frequency ω_c, and rejects higher frequencies; it is said to have a bandwidth equal to ω_c.
- A high-pass filter passes frequencies higher than its cutoff frequency ω_c, and rejects all frequencies below the cutoff frequency; it has an infinite bandwidth.
- A bandpass filter passes a band of frequencies, $\omega_1 \le \omega \le \omega_2$, and rejects all frequencies lower than ω_1 and higher than ω_2; its bandwidth is equal to $\omega_2 - \omega_1$.
- A bandstop filter stops a band of frequencies, $\omega_1 < \omega < \omega_2$, and passes all frequencies lower than ω_1 and higher than ω_2. Bandstop filters are sometimes referred to as *bandreject*, or *notch* filters. Notch filters are special bandstop filters designed with a narrow response characteristic to stop a particular frequency, called the *notch frequency*.

Note that the high-pass filter is the complement of the low-pass filter, and the bandstop filter is the complement of the bandpass filter. The frequency band(s) that the filter passes is called the *passband*, and the frequency band(s) that is rejected is called the *stopband*.

Gain and Loss Functions

As mentioned previously, the magnitude response of an LTI system provides the gain of the system as function of frequency. For an LTI filter, the *gain function* $G(\omega)$ (measured in dB) is given by

$$G(\omega) = 20\log\left|H(j\omega)\right| \qquad \textbf{(18.40)}$$

where log is the base 10 logarithmic function. However, a filter is often characterized by the amount of attenuation, or loss, it provides. The *attenuation* or *loss function*, $A(\omega)$, is defined as the negative of the gain function:

$$A(\omega) = -20\log\left|H(j\omega)\right| \qquad \textbf{(18.41)}$$

Passive RLC Filters

Passive filters are made up only of passive components that are resistors, capacitors, and inductors. Since these filters do not comprise any amplifying element, they require no internal power supplies. Additionally, they perform well at high frequencies since they are not restricted by the bandwidth limitations of active elements. However, for low-frequency applications, the required inductors are quite large and bulky, which makes their realization difficult.

Any filter—whether it is low-pass, high-pass, bandpass, or bandstop—can be constructed using a low-pass prototype that is normalized to a cutoff frequency (passband edge) of 1 rad s^{-1} with a load resistor $R_p = 1\ \Omega$. The use of normalized values is common in filter design. An n-th order passive filter can be realized using one of the standard topologies.

Active Filters

Active filters eliminate the need for inductors that are expensive, bulky, and are the least ideal of the basic circuit elements. In addition, they can improve gain accuracy and frequency response, while allowing for tuning. When comprising resistors, capacitors, and active devices (e.g., operational amplifiers), they are referred to as "Active RC Filters." Any low-power passive filter can be replaced by an equivalent active RC filter. The input or output impedance of an active RC filter can easily be made high or low, allowing for an easy cascade connection of block functions. Using the cascade approach, any transfer function $H(s)$ can be realized using cascaded active RC circuits. The targeted transfer function is first expressed as a product of first-order and/or second-order terms. Each individual term is then realized by a given active RC circuit. Provided that each successive active RC stage does not load the preceding one, a cascade connection of these individual active RC circuits implements the targeted transfer function. A transfer function can be factored into the general form, given as

$$H(s) = K\frac{(s-z_1)(s-z_2)\cdots(s-z_m)}{(s-p_1)(s-p_1)\cdots(s-p_n)} \qquad \textbf{(18.42)}$$

where each zero z_i and pole p_i can be either a real or a complex number. This equation can be rewritten as

$$H(s) = \prod_{i=1}^{q} K_i \frac{a_i s^2 + b_i s + c_i}{e_i s^2 + f_i s + g_i} \qquad \textbf{(18.43)}$$

In this form, each term of the product represents either a real root or a pair of complex-conjugate roots. If $a_i = e_i = 0$ and $b_i = f_i = 1$, then the ith term of the product (i.e., Equation 18.42) will be a first-order term:

$$H_i(s) = K_i \frac{s + c_i}{s + g_i} \qquad \textbf{(18.44)}$$

which has a single real pole and a single real zero. For the case of $a_i = e_i = 1$ in Equation 18.44, the i-th term will be a biquadratic term referred to as a biquad:

$$H_i(s) = K_i \frac{s^2 + b_i s + c_i}{s^2 + f_i s + g_i} \tag{18.45}$$

where $b,c,f,$ and g are real numbers. The bilinear transfer function can be realized either with an inverting op-amp configuration or with a non-inverting one, as shown in Figure 18.11. For each configuration, the design problem involves finding impedances Z_1 and Z_2 assuming that the specifications are the values K, c and, with the restriction that impedances can only be a combination of resistors and/or capacitors.

IC Active Filters

Active filters are increasingly being realized in a monolithic IC form. The all-IC active filter, integrates all filter parts on a single silicon chip. IC active filters are thus significantly smaller in size. However, resistors in monolithic ICs have the disadvantages of requiring large amount of costly silicon area. Additionally, common integrated resistors can exhibit high tolerances and large temperature coefficients that can be too large to be acceptable for the design specifications. There are mainly three different approaches to avoid the use of integrated resistors in the monolithic integration of active filters: MOSFET-C, G_m–C, and switched-capacitor filters.

MOSFET-C filters are similar to fully differential active RC filters, except that resistors are replaced with equivalent MOSFET components.

Switched-capacitor (SC) filters comprise switches, capacitors, and op-amps. Their operation is based on the realization that a capacitor switched at high speed between two circuit nodes is equivalent to a resistor connecting these two nodes. If the switching occurs at sufficiently high speed, the process can be thought of as almost continuous, and an equivalent resistance can be defined as

$$R_{eq} = \frac{(V_A - V_B)}{I_{av}} = \frac{T}{C} = \frac{1}{Cf} \tag{18.46}$$

The impedance of the SC is thus a function of the switching frequency f of the clock signal controlling the switching. SC filters can thus be implemented in digital circuit process technology since large resistors can be simulated by capacitors having small capacitance values. Using this principle, an active RC integrator can be transposed to an equivalent SC integrator as shown in Figure 18.12a. A pair of complementary SC integrators can be used to realize a switched-capacitor biquad filter, as illustrated by the block diagram of Figure 18.12b.

DIGITAL SIGNAL PROCESSING

Digital signal processing (DSP) is a technique of processing signals once they are represented in digital domain by a sequence of numbers or symbols. DSP is extensively used in process control systems, automation, sensor array systems, communications, image processing, medical applications, and so on.

Digital signals are arbitrary binary data streams but are not considered as signals. However, they represent analog signals, which are obtained from physical quantities. DSP is realized by suitably developed algorithms. The algorithms can be arranged to run on the following:

1. Standard computers
2. General-purpose microprocessors
3. Stream processors
4. Digital signal processors
5. Application-specific integrated circuits
6. Microprocessor field programmable gate arrays
7. Industrial digital signal controllers, etc.

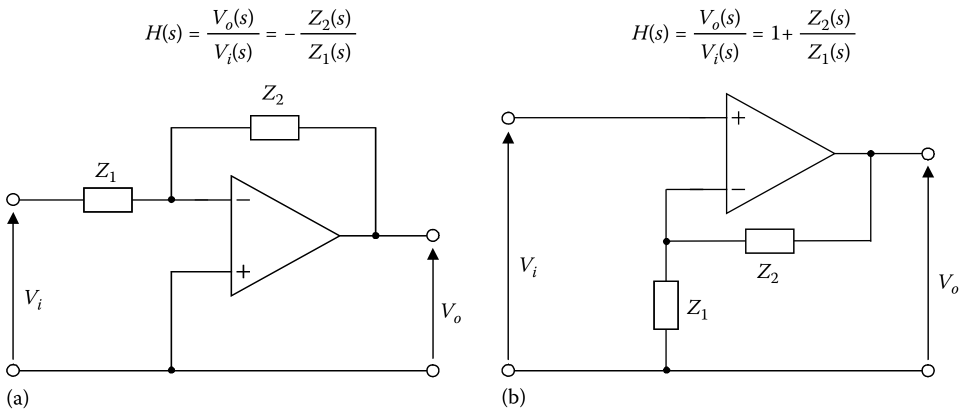

$H(s) = \frac{V_o(s)}{V_i(s)} = -\frac{Z_2(s)}{Z_1(s)}$
Z_2
Z_1
−
+
V_i
V_o
(a)
$H(s) = \frac{V_o(s)}{V_i(s)} = 1 + \frac{Z_2(s)}{Z_1(s)}$
+
−
V_i
Z_2
Z_1
V_o
(b)

FIG. 18.11
Active RC realization of a first-order function for (a) inverting and (b) non-inverting configurations.

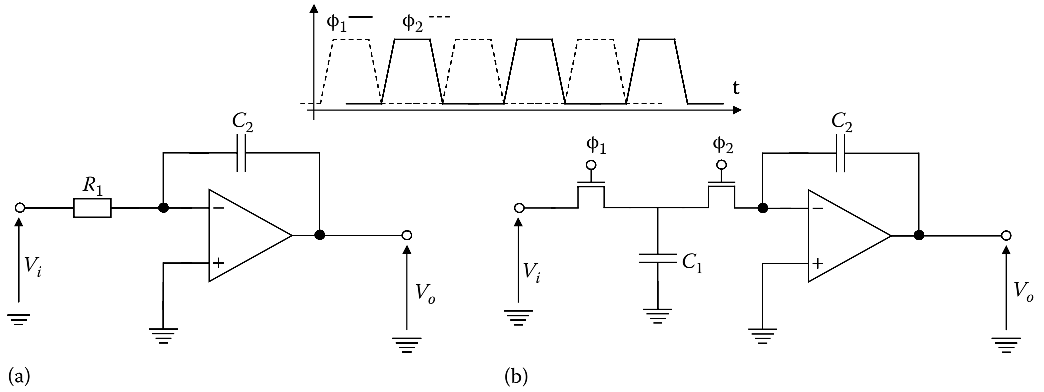


FIG. 18.12
(a) Active RC integrator and (b) its switched capacitor equivalent. Clock signals ϕ_1 or ϕ_2 have a period $T = 1/f$.

Depending on the nature of the signal and experience of the engineers, the digital signals can be analyzed in a number of ways, in the following:

1. Time domain for 1D signals
2. Spatial domain for multidimensional signals
3. Frequency domain as discrete fourier transform and frequency spectrum
4. Autocorrelation and cross-correlation domains
5. Wavelet domain

The signals acquired from the physical variables are usually in analog form. Therefore, traditional digital instruments involve two distinct processing techniques: analogue and digital. Hence, digital signals contain more parts and components compared to analog counterparts. In the conversion process, multiplexers, sample and hold devices, and analog-to-digital (A/D) converters are used. In recent years, many modern digital instruments are manufactured in the form of sophisticated microinstruments. The microinstruments integrate all the essential components of an instrument in a single chip.

A typical digital instrument consists of five main subsystems:

- The analog front end for signal generation and conditioning
- The general-purpose programmable digital hardware
- The storage and communication components
- The application-specific I/O components
- Other ancillaries such as the displays and power supplies

Figure 18.13 illustrates the block diagram of a typical microprocessor-based digital instrument. Many of the microprocessor-based instruments available in the marketplace can implement complex and sophisticated measurements, can store vast amount of data, display the information graphically, and also can function as a part of a measurement system.

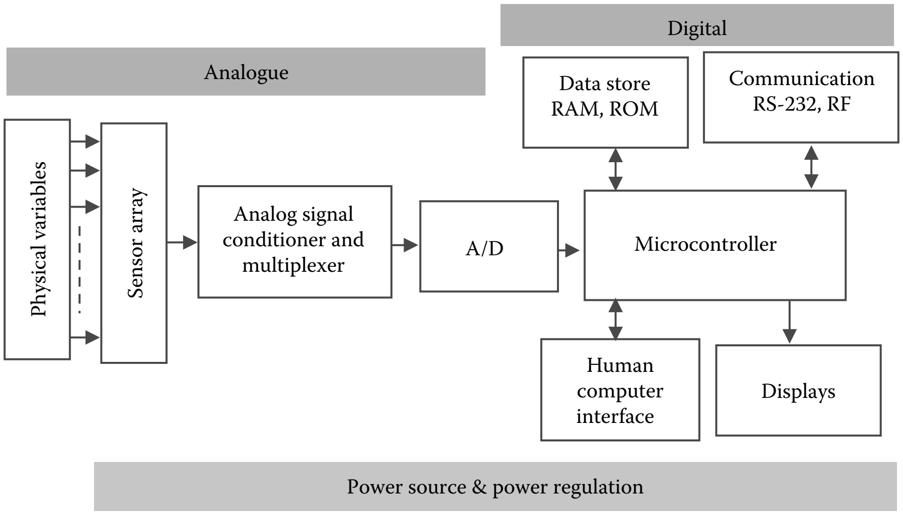


FIG. 18.13
A typical microprocessor-based instrument.

Modern digital instruments are particularly useful in performing complex operations, storing and transmitting information, and communicating the relevant data. This is due to the fact that these instruments are based on digital processors, which provide powerful and flexible signal-processing and data-handling capabilities. Signal processing and data handling can be achieved simply by software and/or firmware implementations. Another important point in digital instruments is that once the signals are converted into digital forms, they can be managed, shared, and processed by other digital devices.

A/D and D/A Converter

Conversion is realized by A/D converters and digital-to-analog (D/A) converters. A/D and D/A converters provide interface between analog and digital environments for computation and data processing purposes. Therefore, they form a vital link between the real-time happenings and the digital environment.

All modern A/D converters produce digital output values in the binary codes that the digits assume values of either 1 or 0. The length of the output word defines theoretical resolution of the A/D converter and the range of digital values that an A/D converter can produce. The smallest change in the input voltage that A/D can detect is

$$V_{LSB} = \frac{V_{\max} - V_{\min}}{2^n} \tag{18.47}$$

where

n is the number of bits produced by the converter

$V_{\max}$ and $V_{\min}$ are the maximum and minimum input voltages that an A/D converter can process correctly

There are many different types of A/D converters available commercially. The reason for this is that there are diverse ranges of applications exhibiting a broad range of signal characteristics. The most popular A/D converters are counter ramp, successive approximation (serial), flash (parallel), and Delta-Sigma (DS) converters. These converters are discussed in detail in chapter 19 of this book.

MATHEMATICAL APPROACH TO DIGITAL SIGNAL PROCESSING

Analog signals are continuous in time and magnitude; sampled analog signals are discrete in time and continuous in magnitude. Digital signals are discrete both in time and magnitude. Digital signals can either be generated by software or derived directly from analog signals using A/D converters.

Mathematically, signals are represented as functions of one or more independent variables. For example, time-varying voltage and current signals obtained from many sensors are functions of single variable (i.e., time), the EM signals are functions of time and space in x, y, and z in Cartesian coordinates. Often, signals from sensors need to be analyzed and processed to the extract useful information. Among many others, the *Fourier transforms* and *Fourier series* are two common methods that can be used in signal analysis. In these methods, the signal representation involves decomposition in terms of sinusoidal or complex exponential components. The decomposition is then said to be in *frequency domain*. The decomposition is determined by *frequency analysis tools* that provide a mathematical and pictorial representation of frequency components, which is referred as the *frequency spectrum*. The frequency spectrum can be determined by *frequency* or *spectral analysis* techniques. The Fourier series is useful in representing continuous-time periodic signals, whereas Fourier transforms are useful in representing periodic signals as well as non-periodic signals, which are also known as *finite energy* signals.

Signals are classified according to the nature of the independent variable. If the independent variable is continuous, the corresponding signal is said to be a continuous-time signal and is defined for a continuum of values of the independent variable. An example of continuous signal is the sinusoidal signal and the sine wave can be used to explain the fundamentals behind the continuous signal processing. A sinusoidal signal $x(t)$ can be described by

$$x(t) = A\sin(\omega t + \theta), \quad -\infty < t < \infty \tag{18.48}$$

where

A is the amplitude

$\omega(2\pi f)$ is the frequency in rad/s

θ is the phase in rads

By using Euler identity ($e^{\pm j\Phi} = \cos\Phi \pm j\sin\Phi$), the relationship can be expressed in exponential form of

$$x(t) = A\sin(\omega t + \theta) = \frac{A}{2j}e^{j(\omega t+\theta)} - \frac{A}{2j}e^{-j(\omega t+\theta)} \tag{18.49}$$

This indicates that a sinusoidal signal can be obtained by adding two equal amplitude complex-conjugate exponential signals. The use of complex exponential tools plays a major role in the signal analysis.

Discrete-time sinusoidal signal may be expressed in terms of sequences as

$$x(nT) = A\sin(\omega nT + \theta), \quad -\infty < t < \infty \tag{18.50}$$

where

n is the integer variable termed as the *sample number*

T is the time interval between samples known as *sampling period*

The *Discrete Fourier Series* representation of a periodic discrete-time signal $x(n)$ sequence is

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} c_k e^{j2\pi kn/N} \qquad \textbf{(18.51)}$$

where N is the fundamental period (repeats itself in every N intervals). The coefficient c_k is given by

$$c_k = \sum_{n=0}^{N-1} x(n)\exp\left[\left(\frac{j2\pi}{N}\right)(-kn)\right] \qquad \textbf{(18.52)}$$

The Fourier coefficients c_k can be calculated as a sequence of numbers for

$$n = 0, 1, 2, 3, \ldots, N-1 \quad \text{and} \quad k = 0, 1, 2, 3, \ldots, N-1$$

Sampling operation obtains a signal, which is discrete in the time domain. The sampling of a signal consists of multiplying itself by a train of impulses separated by the sampling period T. The sampled signal is also a train of impulses, modulated in amplitude. Figure 18.14 shows the sampling process of a signal $x(t)$, Figure 18.14a, using a train of impulses $p(t)$, Figure 18.14b, to obtain the sampled signal $X_s(t)$, Figure 18.14c.

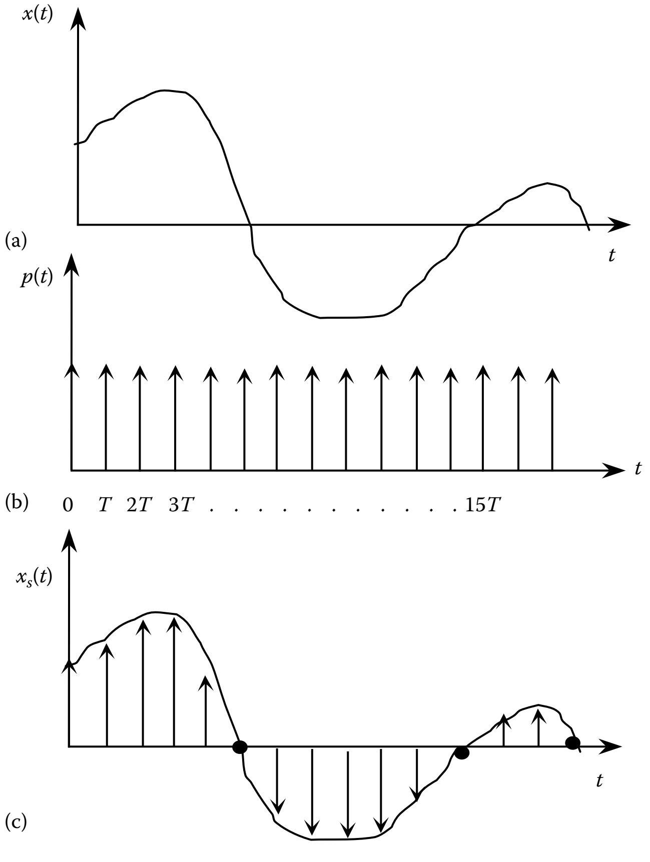


FIG. 18.14
Sampling process of a signal x(t). (a) continuous signal, (b) pulse train, and (c) sampled signal.

Sampling operation can mathematically be expressed as

$$x_s(t) = x(t)p(t) \qquad \textbf{(18.53)}$$

where $p(t)$ is periodic impulse train that can be described by

$$p(t) = \sum_{-\infty}^{\infty} \delta(t - nT) \qquad \textbf{(18.54)}$$

where

t is time
T is a sampling interval
n (is a sample number
δt is a unit sample such that $\delta t = 1$ and $0\delta t = 0$ for $t \neq 0$

The sampling theorem allows us to completely reconstruct a band-limited signal from instantaneous samples taken at a rate $\omega_s = 2\pi/\text{T}$. Using above equations, the relationship between continuous analog signal and sampled analog is given by

$$x_s(t) = x(t)\sum_{-\infty}^{\infty} \delta(t - nT) \qquad \textbf{(18.55)}$$

Example

Take an analog signal $x(t) = A\sin(\omega \text{t}) = A\sin\ 2\pi \text{ft}$, where $-\infty < t < \infty$. After sampling, substitute $t = nT$.

$$x(nT) = A\sin(2\pi f nT) = A\sin(n\theta)$$

where $\theta = 2\pi fT = 2\pi f/f_s$ and $T = 1/f_s$.

θ is called the digital frequency and it can only be between 0 and π. Since $0 \le f < f_s$, hence $0 \le \theta < \pi$.

Because lowering sampling rate saves the processor memory and computational time, there is a strong incentive to set sampling frequency as low as possible, while maintaining faithful and accurate representation of the original signal. The sampling theorem of DSP sets the lower limit on the sampling frequency of a band-limited signal and specifies a method for the recovery of the original analog signals.

Digital Filters

DSP has capability of performing any function that analog signal processing can do. However, filtering is probably the most common DSP technique that is used. It removes unwanted signal components, such as distortions or noise, or alternatively, selects only the desired components, such as selection of a particular frequency band. As a system, a filter can be described by several key characteristics, such as frequency, phase, and impulse responses.

All practical filters, whether analog or digital have at least three regions, a "pass" band, the region of the frequency spectrum that is passed through the filter without significant change, the "stop" band, the region that is not passed through, and a transition region between the pass and stop bands. Existing terminology divides all filters in four basic types, the *low-pass*, *high-pass*, *bandpass*, and *bandstop*. Frequency responses of each filter are specified in terms of their cutoff frequencies, the required accuracy in passing desired frequencies, and the degree of attenuation of unwanted ones.

The two most common digital filters are the *Finite Impulse Response* (*FIR*) filter and the *Infinite Impulse Response* (*IIR*) filter. Both of these filters can easily be implemented by microprocessor systems and DSPs. For LTI systems, the FIR filter computes its output samples as a weighted sum of delayed input samples:

$$y(n) = \sum_{k=0}^{k=N} h(k)x(n-k) \qquad \textbf{(18.56)}$$

where

$x(n)$ is input signal

$h(k)$ is a set of coefficients that define the filter performance

As an example, a 5 coefficient filter $y(n)$ can be expressed as

$$y_5 = h_0x_5 + h_1x_4 + h_2x_3 + h_3x_2 + h_4x_1 \qquad \textbf{(18.57)}$$

Computational implementation of the FIR filters is accomplished with a set of memory elements to store $x(n - k)$ samples and an add–multiply arithmetic unit. The FIR filter coefficients $h(k)$ also forms its impulse response.

The calculations are performed for each sample of the input signal, shifting newly sampled values into the memory, recalling coefficients from the memory and outputting the calculated values to appropriate locations.

IIR filters are also called a recursive filters, because they use both input and previously calculated output values to compute current output value. Like the FIR filters, the IIR filters also employ memory elements and perform add–multiply operation. The IIR filters are digital equivalents of the analog filters and, for the comparable performances, they require fewer calculations than the FIR filters. In both cases, if the signal has periodic, then the frequency response $H(e^{j\theta})$ can be written as

$$H(e^{j\theta}) = \sum_{n=-\infty}^{\infty} h(n)(e^{-jn\theta}) \qquad \textbf{(18.58)}$$

For sampling spacing T, $h(n)$ can be expressed as

$$h(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H(e^{j\theta})(e^{jn\theta})d\theta \qquad \textbf{(18.59)}$$

This coefficient is used in many filter applications.

Example: Determine a Low-Pass Digital Filter Coefficients

We want the filter to pass frequencies from $-\theta_c$ to $+\theta_c$, hence $H(e^{j\theta}) = K$:

$$\begin{aligned} h(n) &= \frac{1}{2\pi}\int_{-\theta_c}^{\theta_c} H(e^{j\theta})(e^{jn\theta})d\theta \\ &= \frac{1}{2\pi}\int_{-\theta_c}^{\theta_c} K(e^{jn\theta})d\theta = \frac{K}{2\pi}\left[\frac{e^{jn\theta}}{jn}\right]_{-\theta_c}^{\theta_c} \end{aligned}$$

Giving the coefficient for a low-pass filter as

$$h_{LP}(n) = \frac{K}{2n\pi}\left[\frac{e^{jn\theta_c} - e^{-jn\theta_c}}{2j}\right] = \frac{K}{n\pi}\sin(n\theta_c) \qquad \textbf{(18.60)}$$

where θ_c is the cut off frequency and $n = 0, \pm1, \pm2, \pm3, \ldots, I$, and $I = (N - 1)/2$.

Once the coefficients are determined, they need to be shifted to make them *causal*. Similarly, coefficients for other filters can be determined, where

High-pass filter

$$h_{HP}(n) = -h_{LP}(n) \qquad \textbf{(18.61)}$$

Bandpass filter

$$h_{BP}(n) = 2\cos(\theta_0 n)h_{LP}(n) \qquad \textbf{(18.62)}$$

where $\theta_c = (\theta_u - \theta_l)/2$ and $\theta_0 = (\theta_u + \theta_l)/2$.

Bandstop filter

$$h_{BS}(n) = (-1)h_{BP}(n) \quad \text{and} \quad h_{BS}(0) = K - h_{BP}(n) \qquad \textbf{(18.63)}$$

DIGITAL SIGNAL PROCESSING SOFTWARE SUPPORT

Signals produced by the transducers and sensors need to be conditioned before input to A/D converters. Process signal conditioning is appropriate when

1. There are a large number of channels needing to be multiplexed
2. Mix I/O and transducers are used
3. Operating in noisy environment

4. Signal levels are low requiring amplifiers
5. Signal needs to be isolated from the digital environment
6. Unwanted noise exists in the signal
7. Filtering is necessary to extract useful information
8. Linearization is required

Once the signal is at the port of the data acquisition hardware, software needs to be used to convert the signals into digital form. Data acquisition software is usually provided by the manufacturers of the data acquisition equipment and cards. Conversion requires basic steps, such as

- Identification of the channel to acquire data from
- Using library functions to start acquiring
- Retrieving acquired information from the channel
- Save, plot, or display the data on the user interface
- Stop data acquisition
- Relinquish the resources created for that channel
- Scan other channels if required by setting scan rates and number of scans

Similarly, digital inputs and outputs are carried out by

- Reading from the digital lines
- Reading from the digital ports
- Writing to digital lines
- Writing to digital ports

Once the signals are represented in binary form and placed in memory space, the signals can be

- Filtered as FIR or IIR filters
- Arithmetic and logic operations can be conducted
- Modulated or demodulated
- Transmitted

As an example, some of the software commands offered by MATLAB® are given in Table 18.2. Two examples will be given to illustrate DSP.

TABLE 18.2
Some Functions in MATLAB for Signal Processing

abs	Computes absolute value
conv	Multiplies two polynomials
cos	Computes cosine
exp	Computes the exponential
filter	Filter data
inv	Inverts the matrix form
log 10	Computes natural logarithm
max, min	Takes maximum or minimum values
roots	Determines the roots of a polynomial
sign	Implements signum function
sum	Determines the sum of elements in vector

Example: Design of a Tunable Causal FIR Low-Pass Filter

```
% Program 1.1
% Tunable FIR Filter Design
clf;
w=0:pi/255:pi;
f=[0 0.36 0.46 1];m=[1 1 0 0];
b1=remez(50, f,m];
h1=freqz(b1,1,w);
m1=20*log10(abs(h1) );
n=–25: –1;
c=b1(1:25)./sin(0.41*pi*n);
wc=[0.31*pi 0.51*pi];
for i=1:2
    d=c.*sin(wc(i)*n);
    q=(b1(26)*wc(i) )/(0.4*pi);
    b2=[d q flip1r(d)];
    mag(I,:)=20*log10(abs(freqz(b2,1,w)));
end
Plot(w/pi,mag(1,:),‘b—‘,w/pi,m1,’r-‘,w/pi,mag(2,:),’g-’);
…
```

Example: Plot Discrete-Time Signal sin(nπ/8) and Find Its Mean

```
% Program 1.2
n=[0:32];
x1=sin(pi*n/8);
yi=mean(x1);
stem (n,x1)
title(‘x1=sin(pi*n/8)’)
xlabel(‘n (sample)’)
ylabel(‘x1[n]’)
```

DSP Software Development

DSP can be divided into three major categories:

1. Processing of signals on real-time basis as the events take place
2. Off-line processing for analysis and synthesis
3. Programming of DSP chips for embedded systems

Developing signal-processing systems generally takes place either by using hardware only, or the software only, or combinations of hardware and software. The software comes in two forms: firmware and high-level language algorithms, which runs on a processor. The firmware approach is essentially a hardware approach, which allows the user to dynamically reconfigure the hardware processor based in FPGAs at different points in the processing.

The DSP implementations are historically dominated by hardware driven by the general purpose software for sensor signal-processing, instrumentation, automation, motor control, motion control, PID loops, etc. The hardware consists of general purpose microprocessors or computers of dedicated DSP chips. A DSP is a *programmable* device, with its own set of instruction code capable of carrying out millions of floating point operations per second. DSPs are usually embedded within complex "system-on-chip" devices, analog and digital circuitry. For more information, readers can refer to Chapter 19.

Programming languages (i.e., Assembly, C, C++, Pascal, Visual Basic, Fortran) can be used for developing algorithms for DSP. For this, programming languages need to have the following basic elements:

- Organizing different types of variables and data
- Describing the operation to be done
- Controlling the operations based on the results obtained
- Organizing the data and operations to execute a sequence of programs
- Enabling the movement of data to outside world and program

DSP Software Packages

There are numerous packages such as Verilog HDL, Simulink, Maplesoft, Mathcad, LabView, and many others.

In addition, many PLCs and SCADA systems have their own signal-processing capabilities and they provide analog input and output modules for continuous signals.

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